SYSTEM AND METHOD FOR PROVIDING IP/INTERNET TELEPHONY

FIELD OF THE INVENTION

The present invention generally relates to a system and method for providing internet telephony. In particular, the present invention relates to a system and method of providing a wireless internet telephone system over either a regular dial up telephone or a cable network.

BACKGROUND

One of the primary reasons for interest in offering Internet Protocol (IP)/internet telephony services is the pricing structures currently in place for the data service, and voice service offered by telephone operators. Long distance voice service can be thought of as "demand data" service, where the user pays a premium for the instantaneous access to a 64 Kbps channel (voice grade channel in the US). Widely publicized, promotional type pricing for this service is on the order of $.10 a minute. By contrast, data service offered by telephone operators, such as that offered for a T-1 connection (24 voice quality data lines, for a 1.544 Mbps connection) is priced at approximately $1000/month, which works out to $.001 a minute per voice line. In the very near future, cable operators will place extreme pressure on even the data service rates for telephone operators, as cable modem will allow cable operators to offer hundreds of Kbps effective throughput for approximately $50/month.

The basic idea of IP/internet voice telephony is to digitize your voice as you talk on the phone and send the digitized data as IP packets to the Internet. An IP voice device can be embedded within an Internet connection device such as a modem, a set-top-box, or a computer. It can be also built as a stand alone product. The stand alone IP voice device, for example, may provide an Ethernet jack which can be connected with an Internet connection device and other LAN devices. The IP voice device may also include interfaces to connect regular phone handsets. The quality of speech heard through a normal telephone line requires 64kbits/s bandwidth. However, most current internet connections have less bandwidth, such as 28.8 Kbps, or 56 Kbps modem. Furthermore, even if a fast connection device is

used, such as an ISDN, or Cable Modem, the Internet network itself is a shared medium and has limited bandwidth. Therefore, audio codecs are usually embedded to compress the voice data.

To guarantee the interoperability between IP voice devices from different vendors, the International Telecommunications Union (ITU) developed H.323 as the standard for telephony over IP network. H.323 defines common procedures for call setup, data compression, and data transport.

In a general sense, IP telephony can be thought of as providing a "virtual" point to point connection for voice services over Internet. An IP voice device is basically a gateway to connect the regular telephone system to the Internet. The following example demonstrates how a call would be placed. A user in Indianapolis wants to call a friend in Paris. He picks up his IP voice device handset (or activates a virtual handset on a computer screen for a "built-in" version) and hears a dial-tone like a regular telephone dial-tone. Then, he dials his friend's Paris phone number. The call travels over the Internet to some Switching Server provided by the IP telephony service provider. The Switching Server will connect the call to his friend's IP voice device and initiates the call. If his friend has only a regular telephone, the Switching Server will connect the call to a gateway in Paris. The gateway in Paris then initiates a call over the public switched telephone network (PSTN) to the local Paris number. The cost to make phone calls between Indianapolis and Paris using two IP voice devices is only the Internet access fee. If one party uses a regular telephone, the extra charge is merely that of a local dial call.

Depending on the Internet connection, there are at least two methods for making calls using an IP voice device: dial-up connections, and direct connections. With a dial-up connection, a user first calls an ISP (Internet service provider) over a regular dial-up line to set up an Internet connection. Then, he will use the IP voice device handset to dial the phone number of the person he is calling. The present applicants recognize one problem with this approach is that the recipient must be online waiting for the call. So, the sender may have to first call the recipient using a regular phone to make the appointment. With a direct connection, a user places a call using the IP voice device just as he does with the regular telephone. The direct connection indicates a permanent open channel to the Internet such as ISDN, or a cable access device. For a

a dial-up connection call, a phone that has been called won't ring unless the Internet connection is already established for this phone. For a direct connection call, a phone would ring like a normal telephone.

The are many advantages to IP/Internet telephony. One such advantage is reduced cost as described above. A low bit rate audio codec embedded in the IP voice device enables voice calls over a 28.8 Kbps modem. For a small reduction in voice quality, a person's monthly phone bill will be greatly reduced. If IP voice device used together with a cable modem, the private service network plus high bandwidth of the cable modem will provide very good sound quality. Even if the voice quality provided by a IP/internet voice device is unsuitable for all phone communications, a IP/internet voice device may be useful as a second-line residential phone. Also, the H.323 standard supports several well defined conference modes and, therefore, IP voice device is able to be used for multi-point conference calls. A "Web" dial-in service is advantageous for technical or customer support lines because, for example, an Internet address of a company's IP voice device can be embedded in the company's Web page and customers can then call the company simply by "clicking on" that Internet address. The cost associated with toll-free ("800" number) telephone numbers will be reduced as a result.

In addition, MSOs (cable television system operators) have recently become interested in adding inexpensive telephony services using a combination of an MSO's private HFC (Hybrid Fiber Coax) network and the public Internet. Voice signals are converted to digital values and transported across the networks using various established and proposed Internet protocols as IP (internet protocol) packets.

Reference D1 (WO 97 29581 A) discloses a transmission system which enables users to have a voice conversation via the internet. D1 utilizes the PSTN to connect with an internet service provider. The system disclosed by D1 includes transmitting a telephone call to the PSTN and an originating voice engine which compresses the signal for transmission over the internet. The

signal is then transmitted over the internet to a receiving voice engine. The receiving voice engine decompresses and demodulates the received signal and provides the signal to the PSTN (31). The PSTN compresses the signal into a format for transmission therealong for receipt by a receiving telephone. This system compresses and decompresses a signal into a format suitable for transmission by the PSTN. Furthermore, the compression and decompression of the signal is performed at the PSTN, remotely from the ends of the established communication channel. Thus, although this system eliminates most long distance charges associated with a voice call, there are still local charges associated therewith and possibly long distance charges on either side of the communication channel associated with contacting the internet service provider.

Reference D2 WO 98 11703 was published 19 March 1998.

However, there are also problems associated with existing IP/Internet telephony systems. For example, the above-described systems involve some combination of additional or revised POTS (Plain Old Telephone System) wiring, additional or revised cable network wiring, or additional network interface boxes. In addition, any connection which replaces a PSTN (public switched telephone network) service (such as reuse of the existing POTS wiring within the home to replace PSTN services with HFC telephony services) may be required to supply so called "life-line" services. Some of these options require professional installation which may be costly, time consuming, and inconvenient for the user.

SUMMARY OF THE INVENTION

The invention resides, in part, in recognition of the above-described problems and, in part, in providing a system and method for solving these problems. In particular, the inventors recognize that the described problems are solved by providing a voice call over an Internet connection by receiving a signal from a cable network. The signal represents internet protocol data packets of the voice call and is both modulated in a first format and compressed to match a format of the cable network. The signal is demodulated and decompressed. The signal is next compressed into a format of a home environment, modulated into a

second format and wirelessly transmitted to a wireless device. The signal is then demodulated and decompressed in the wireless device. The inventors also provide a system including the elements necessary to carry out this method.

An aspect of the present invention involves providing an internet telephony system using a wireless connection such as via the unregulated 900 MHz cordless phone spectrum or other spectrum allocated for wireless communications to provide an RF link between an IP connection device, a network interface box or set-top box; and one or more wireless handsets. A processing/control element in the network interface box would run the required IP protocols to establish and manage call set-up and teardown (currently defined within ITU-T H.323), translate the digital voice signal between IP and the local RF link protocol, and provide the RF base station function for the handset(s). Each handset would incorporate the other end of the RF link, and A/D and D/A functions to convert the voice signal to and from digital packets, and potentially apply some compression algorithm to improve bandwidth utilization. In a handset design which does not incorporate enough processing power to perform the compression function, this function could potentially reside in the network interface box.

Another aspect of the present invention involves a mechanism to establish a wireless interface to a telephone handset through a settop box that is tied into a cable network such as a hybrid coaxial cable network.

Another aspect of the present invention involves using a standard protocol such as the Internet Protocol to maintain a digital connection into a cable network while using an RF link to transmit compressed voice/data information between a telephone device such as a telephone handset and an interface unit such as a settop box.

Aspects of the present invention also involve providing for eliminating the need to add wiring, such as POTS wiring, to accommodate one or more handsets, or alternatively eliminating the need to add multiple cable drops and adapters such as POTS/HFC adapters. A wireless feature in accordance with aspects of the invention provides for coupling a network interface box to an existing cable outlet and for adding handsets as required without installation of additional outlets. In addition, aspects of the invention provide for multi-line Internet phone calls without rewiring. Another aspect of the invention involves adding an analog trunk interface wherein an IP voice device can be connected to a PBX device for providing an Internet PBX. For example, a user could dial a prefix, such as "9" to make a regular outside phone call, or dial a different prefix, such as "8" to make an Internet phone call.

In accordance with another aspect of the present invention, an IP voice device or a set top box provides for connecting to external equipment, such as a PC or Workstation, and utilization of computation power of external devices for providing additional features such as IP FAX service or video conferencing.

BRIEF DESCRIPTION OF THE DRAWING

The invention may be better understood by referring to the accompanying drawing in which:

Figure 1 shows, in block diagram form, an embodiment of a system incorporating aspects of the invention; and

Figures 2 through 7 show, in block diagram form, embodiments of portions of the system shown in Figure 1.

Figure 8 is a flow chart illustrating a method of operation according to the principles of the present invention.

DETAILED DESCRIPTION

In Figure 1, a system constructed in accordance with aspects of the invention comprises a PSTN network and a cable network coupled to a cable modem termination system. The PSTN Network and/or Cable Network provide alternative paths for coupling the system shown to the Internet, e.g., to an Internet service provider (ISP). The cable modem termination system is coupled to a gateway, such as in a home

environment, that comprises a cable modem network interface and first and second codecs for coupling to a conventional wired telephone via a subscriber line interface unit and/or to a wireless telephone unit via an RF modem interface, respectively.

Data transmission between the various units shown in Figure 1 occurs as follows. Data transmission between the PSTN network and the cable modem termination system shown in Figure 1 (path 1 in Figure 1) may occur in 64 Kbps/voice line format or in T1 or higher hierarchy. Data in the cable network (e.g., path 2 between the cable network and the cable modem termination system in Figure 1, or path 2 between the cable modem termination system and the cable modem network interface unit in the gateway in Figure 1) may be carried over TCP/IP compressed at various rates or uncompressed linear at 64 Kbps/voice line. Data transmission between the cable modem network interface and the first codec (path 3 in Figure 1) may occur in linear PCM format at 64 Kbps/voice line. Data transmission between the cable modem network interface and the second codec (path 4 in Figure 1) may occur in linear format at 64 Kbps/voice line or compressed at various rates. Data communication on path 5 in Figure 1 (between the first codec and the subscriber line interface unit) may be in companded format at 64 Kbps/voice line. Data communication via path 6 in Figure 1 (between the second codec and the RF modem interface) may be in linear format at 64 Kbps/voice line or in compressed format at various rates. Data communicated to and from the subscriber line interface unit (path 7 in Figure 1) may occur in analog format (e.g., for an RJ11 connector) and data communicated to and from the RF modem interface (path 8 in Figure 1) may occur in RF digital modulation format.

In embodiments shown in Figs. 2 and 3, IP telephony compression algorithms, call setup, and a cordless telephone adapter are incorporated into an IP connection device or a client server. An example of an embodiment of such a device is a device referred to as a Network Computer (NC) which is a computer similar to a personal computer (PC) that is intended primarily for providing an interface to the Internet. That is, a network computer is intended primarily to provide computing power and features sufficient, for example, to connect to the internet, execute web browser software and provide email capability. A cordless telephone adapter in accordance with aspects of the invention would allow the convenience to call from any room in a house without expensive rewiring.

The phone would ring only when there is an incoming IP phone call, and would present dial tone, etc. when used to place a call.

Two exemplary embodiments of an IP connection device having a cordless phone interface are shown in Figures 2 and 3. The system shown in Figure 2 utilizes an analog cordless telephone interface such as CT-1 (46/49 MHz). The system shown in Figure 3 uitlizes a digital 900 MHz spread spectrum cordless telephone interface. The analog cordless IP voice device may provide a lower cost solution. However, a digital 900 MHz cordless IP voice device may be more advantageous in terms of voice quality and expandability. For example, a digital cordless phone typically provides better voice quality due to the noise cancelling capability of the digital system and a digital cordless IP voice device may have more than one handset. Also, a cordless IP voice device such as that shown in Figure 2 and/or 3 may be used for data service when used together with wireless modem.

The systems shown in Figures 2 and 3 may include a voice codec for compressing and decompressing the voice data if the modem of the IP connection device is running at low speeds. Table 1 lists some popular standard voice codec algorithms and their associated data rates.

Standard	Data Rate
G.711	64 kbps
G.723.1	5.3/6.3 kbps
G.728	16 kbps
G.729	8 kbps
GSM	13.3 kbps

Table 1 Voice compression standards

Figure 2 shows a client server device including IP voice features which, for example, may be included on an IP voice adapter card included in the client server device. The IP voice feature includes a CT-1 subsystem comprising RF transmitter circuitry Tx and receiver circuitry Rx, a programmable PLL synthesizer, a baseband (audio) processor, and a microprocessor interface. The components Tx and Rx and the PPL synthesizer are used to modulate and demodulate RF signals for transmission and reception of the wireless telephone signals. A duplexer is used to separate the transmit and receive paths

of the RF communications. As discussed before, a PCM codec integrated with filters may be needed to provide A/D and D/A conversions and compression, as well as the transmit and receive filtering of the signals. The digital signal processing (DSP) unit may be, for example, an integrated circuit (IC) that implements the voice codec under the control of the CPU (central processing unit) which may be a microprocessor. The CPU provides the central control of the wireless IP interface device shown in Fig. 2. The CPU is connected to the various components of the device via a data control bus. The CPU has a built in memory for storing the required control codes, including implementation of the H.323 standards and the TCP/UDP/IP protocols.

Figure 3 shows another exemplary IP connection device having a digital cordless phone interface such as a 900 Mhz interface. A baseband device usually includes a spread-spectrum modem, an audio engine (PCM, DTMF, etc.), a voice codec, and a microcontroller. The components Tx and Rx and the PLL synthesizer are used to modulate and demodulate RF signals for transmission and reception of the wireless telephone signals, to and from the wireless handset. A duplexer is used to separate the transmit and receive paths of the RF communications. A DSP unit is used to implement the voice codec under the control of the CPU. The CPU, or central processor, provides the central control of the wireless IP interface device as shown in Fig. 3. The CPU is connected to the various components of the device via a data control bus. The CPU has a built in memory for storing the required control codes, including impelmentation of the H.323 standards and the TCP/UDP/IP protocols.

Another aspect of the present invention is a wireless internet telephony system to be connected to a cable network. A network architecture in accordance with the principles of the present invention is shown in Figure 4.

In Figure 4, an interface to cable network(100) comprises a cable modem termination in the physical layer that has a bi-directional channel connected to the hybrid fiber coax network (105). The physical layer modulation scheme may comprise, for example, Quadrature Amplitude Modulation (QAM). The transport mechanism may comprise TCP/IP. In order to enable the voice application over the cable modem, the network interface unit may employ a protocol such as H.323 over TCP/IP. This enables signaling, call set up and other functions. The voice (fax and

analog modem is included in this paradigm) data itself may be carried in a compressed or uncompressed format. For example, companded voice data at 64 Kbps can be carried over the cable network embedded in TCP/IP packets. Alternatively, it may be compressed using one of many voice compression methods and carried over the cable network embedded in TCP/IP packets. Certain types of data cannot be compressed (example fax or analog modem) and need to be carried in a linear format.

Figure 4 also depicts a wireless interface (104) to a plurality of handsets or receiver devices (101, 102, 103...). The protocol between the base device (100) and the handsets may be entirely proprietary or some standard interface. Additionally, the data format or voice (compressed in one of many possible algorithms or uncompressed) may be different in the RF network as compared to the format in which the voice was carried over the HFC network.

Advantages associated with maintaining the same data format (e.g., compression scheme) in the wired (cable) and wireless network are:

1. only a single encoding/decoding process is necessary which, in a home environment, can take place at the wireless handset or mobile terminal (multiple transcoding processes normally result in degradation of the original source material); and
2. the base station (e.g., in the home) is transparent to the data from the handset or mobile terminal.

An advantage associated with maintaining different data formats (e.g., compression scheme) in the wired (cable) and wireless network is that certain compressed formats are specifically suited to be carried over certain transmission channels. Channel errors, depending on how they occur, can cause different degradation to the source material depending on the compression scheme that is employed. The wired and wireless environments are very different in terms of channel characteristics. Therefore, tailoring the coding scheme to match the characteristics of the channels may have some benefits in the overall system design.

Figure 5 shows further details of an exemplary embodiment of cable set-top box (100) in Figure 4. The cable channel (91) that carries both downstream and upstream data is usually frequency division multiplexed to enable simultaneous channels of operation. Further, within a specific channel, due to the nature of the shared cable medium.

multiple users may signal using a time division multiplexed access mechanism. This task is coordinated by the head end.

The cable interface (40) is a network interface unit (NIU) comprising of a modulator/demodulator pair and a processing unit for interpreting the incoming data stream and messages. One of the transport mechanisms employed is TCP/IP. The NIU receives data, demodulates decodes and extracts the information pertaining to specific voice channels in this application. It is also responsible for maintaining signaling information with the external network (for example using the H.323 protocol stack or any other commonly used signaling stack used in telephony). Additional features such as caller ID, messaging, voice mail etc. are features that are supported by the NIU. This is enabled by its interface with the Caller ID block (50), the external digital signal processor (10) with an embedded microprocessor (5) that coordinates the task of messaging, and voice compression/decompression as necessary.

The incoming messages are stored in compressed or uncompressed format in the message memory(60). Other system architectures may be used wherein the messages are stored in message memory in yet another compressed format to increase the time over which messages can be stored in a given amount of available memory. This task of additional compression/decompression may take place in DSP unit 10. The code memory (70) contains the code for the DSP engine. The RF cordless circuitry (20) is responsible for communicating with the handsets or mobile terminals and exchange specific information intended for each device. In addition to the exchange of data, 20 is also responsible for exchanging signaling and status information. The system shown in Figure 5 includes a common bus (80) between the functional components for data exchange, but a generalized architecture need not be limited to the bus structure shown in Figure 5. Additionally, messaging information and caller ID information are exchanged between 100 and the handsets or mobile terminals through the RF/cordless circuitry.

Figure 6 shows an exemplary embodiment of the receiver/set-top box 100 described earlier in regard to Figure 4 and referred to as unit 700 in regard to Figure 6. The transmit and receive signals into the cable network through the RF connector (796) are kept isolated using a diplexer (795). The cable tuner (705) and demodulator(710) convert the digitally modulated signal (for example QAM) into a composite digital bit stream which is delivered to a Medium Access Control – MAC (720) block

that performs the task of separating information into logical transport streams. Additionally, unit 720 is responsible for synchronizing with the cable head end in order to provide the settop box access control to the common cable medium for return channel information. The burst modulator (740) and power amplifier(730) create and send data in the return channel path back into the cable network.

The RF processing chain for processing the digital information from the cable network starts with the interface, or input/output (I/O) unit (760) which may be implemented as an application specific integrated circuit (ASIC) and which interfaces with a cordless phone processing unit (750), that also may be part of an ASIC or a separate ASIC, to create individual links with handsets or mobile receivers. Unit 750 is coupled to memory units DRAM 765 and ROM 770 for receiving stored processing instructions and for temporary data storage during processing. The data intended for each individual handset or mobile receiver may be time slotted, modulated and sent on the RF link through the RF connector (797). Additional information streams processed by the MAC processing block (720) may be directed to an ethernet port (783) through an ethernet controller (781) or an USB (Universal Serial Bus) port (784) through an USB controller (785) or an RS232 interface (791) through an RS232 driver(790).

The various functions shown in Figure 6 are connected to bus 721 for communication of data and control information between the functions and between the functions and CPU 786 which controls the operation of the functions in device 700. Also coupled to bus 721 are memory units 775 and 780 for storing control programs and data for CPU 786 and other functions in device 700. Power for unit 700 is provided by power supply 792. Also, while many of the processing blocks shown in Figure 6 may be optional depending on the specific application or product, the system shown in Figure 6 illustrates the composite nature of the data coming over the cable system. The path for the voice channels are of particular interest in regard to the present invention.

Fig. 7 shows a block diagram of an implementation of a wireless handset 101. The handset 101 comprises a DSP unit 201 including a microprocessor 210, a speaker, earpiece, RF circuitry and a keypad. The microprocessor 210 controls the various components of the wireless handset 101 via a system bus 202. The RF circuitry is connected to a RF antenna for transmitting and receiving RF wireless signals. A keypad 204

is used for an user to dial a phone number and for controlling other functions of the wireless phone. The DSP converts the analog signal into a digital signal to be transmitted over the RF spectrum if a digital transmission system is used. Memory 203 stores the program codes to be executed by the microprocessor 210.

It is to be understood that the embodiments and variations shown and described herein are illustrations only and that various modifications may be implemented by those skilled in the art without departing from the scope and spirit of the invention.

Figure 8 shows a flow chart illustrating a method of operation according to the principles of the present invention. At step 802, a signal is received by, for example, by the cable interface 40 of unit 100. This signal is demodulated by cable interface 40 into a demodulated signal at step 803. The unit may decide to further compress or decompress this demodulated signal under the control of the DSP unit 10 as described before. The DSP then causes this signal to be further modulated at step 805 by the RF/Cordless circuitry 20. This further modulated signal is then transmitted wirelessly to a wireless unit, for example, as shown in Fig. 7. After receiving this further modulated signal at the wireless unit, the wireless unit then demodulates this signal for completion of the IP voice call.

It is to be understood that the embodiments and variations shown and described herein are illustrations only and that various modifications may be implemented by those skilled in the art without departing from the scope and spirit of the invention.